IP Telephony signalling

Overview
H.323
Next Generation Network (NGN) is the ETSI effort to harmonize packet telephony

The network architecture is layered in a much more strict sense than in case of CSN

Services
- IP Applications
- Virtual Home Environment
- Open Service Architecture

Control
- call control
- session management
- mobility management

Switching
- Transcoding at the edge
- Switching
- Routing

IP Telephony Signaling alternatives

In Terminals Intelligence In Network
SIP
- ascii based
- devil in details
- Adopted by 3G
- Bakeoffs drive vendor interoperability

H.323
- Inherits ISDN
- complex
- still few services
- Widely used
- first working solution

Megaco/H.248/MGCP
- newest
- seems to be quality spec.
- architecture holds promise
- Interoperability?

SIGTRAN works on ISUP over STCP over IP
- many view this as an interim solution!
H.323 is a key standard for packet based multimedia communication

H.323 over: LANs, Enterprise Area Networks, MANs, Intranets, Internets
include dial-up connections and PP-connections over CSN/ISDN with PPP packet transport.

Example networks:
- Ethernet (IEEE 802.3)
- Fast Ethernet (IEEE 802.3u)
- FDDI
- Token ring (IEEE 802.5)
- ATM

MM includes:
- Audio (mandatory)
- Video (opt)
- Data (opt)

Communication = conference or two party call.

H.323 is used in IP trunking

IP network is most often an Intranet, not the public Internet.
Fall-back to PSTN may be used in case of overloaded IP -network.
Makes use of arbitrage due to the regulated high International PSTN tariffs.
Initially (1997-98) a promising business - now does not look too promising.
H.323 provides also comprehensive conferencing services!
H.323 uses H.225.0, H.245 and RTP

H.323 Terminal

G.711 and others

RAS

H.225.0

H.245

RTCP

RTP

UDP

IP

Gateway

Gatekeeper

RAS

ISDN/ Q.931

CCS#7 /ISUP

RTP - Real-Time Transport Protocol
RTCP - RTP Control Protocol

Note: this is an example configuration!

H.323 supports many call modes

- Directly between two H.323 endpoints (no GK)
- Between two H.323 endpoints using a GK
- Many conference types
  - ad hoc multipoint conference (start with 2-party call - expand to conf)
  - broadcast conference (one sender, many receivers)
  - broadcast panel conference (mp conf + bc conf)
  - centralized multipoint conference (trms pp to MCU, MP sends to trms)
  - decentralized multipoint conference (no MCU - all to all coms)
  - hybrid multipoint conference - centralized audio or video
  - mixed multipoint conference (mix of decentralized + centralized modes)
Mixed multipoint conference example

H.323 zone is controlled by a Gatekeeper

- Zone has at least one terminal, MCUs and GWs are optional.
- Zone has one and only one GK.
- Gatekeeper controls access to the network for Ts, GWs and MCUs and provides:
  - address translation
  - gateway location
  - bandwidth management

GK control is not enforced, so this is an intranet solution, i.e. all parties engage in cooperation voluntarily.
Audio and Video coding

- Audio: G.711 is compulsory (PCM –coding). In practice much more efficient coding methods are used based on negotiation.
- Video: H.261 is compulsory. H.263 (from 1995) is mentioned in H.323. Some H.323v4 products claim support of H.264 (= MPEG4 part 10) video coding.
  - H.264 provides DVD quality at 1.1 Mbit/s, cmp to 3 Mbit/s for MPEG2!
  - H.264 is meant for both IP based broadcast/multicast and videoconferencing.
H.323 supports many parallel addressing methods

- H.323 entity shall have at least one Network Address (e.g. IP address)
- TSAP identifiers allow multiplexing several channels sharing one Network Address - map to TCP/UDP port numbers (source port, destination port).
- An endpoint may have one or many Alias addresses - may represent the Ep or a Conference that the Ep is hosting. Include: E.164 numbers, H.323 IDs (e.g. John Smith), e-mail addresses. Aliases are unique in a zone.

Gateway translates between transmission formats, communication procedures and media formats

- Example: H.225.0 to and from H.221 (transm.f)
- H.235 to and from H.242 (comm procedure)
- Media format: Audio, video, data
- Represents characteristics of network endpoint to SCN endpoint and the reverse. May also work as an MCU
- Can also do call set-up and clearing
GK provides call control services, when present, shall do:

- Address translation (e.g. alias to transport address using DNS + E.164 to transport address)
  - uses the translation table produced from registration messages
- Admission control: ARQ/ACF/ARJ of H.225.0
  - based on call authorization, bandwidth, other criteria
- Zone management

GK may optionally do

- Call control signalling. May also direct the endpoints to setup call signalling channel between themselves
- Call Authorization using H.225.0 signalling
- Bandwidth management controls the number of simultaneous calls in the zone
- Call management - keep list of calls -> busy conditions
- GK management, Directory service etc -
Endpoint can discover a Gatekeeper automatically

- GRQ (GK request)
- GCF/GRJ (GK conf/reject)
- [transport address of GK’s RAS Ch, alternateGK, cryptoinfo]

- Automatic discovery eases maintenance of individual terminals
- Terminals may also have the GK id configured

RAS signalling function

- Performs
  - Registration of endpoints, Admission of calls, Bandwidth changes for calls
  - Status
  - Disengagement of endpoints.
- Uses RAS signalling channel /= call signalling channel and H.245 control channel. GKs have a well def. TSAP id for RAS sig. channel
- Endpoint=H.323 terminal or GW or MCU (is callable)
Endpoints register using GK’s RAS Channel
Transport Address prior to any calls are made

Ep \[\text{RRQ} \{\text{Registration rq}\{\text{alias, transport address, …}\}\}
\]
\[\text{GK}\]
\[\text{RCF}\{\text{access token}\}/\text{RRJ}\]

- Security policy may require that registration has time-to-live
  and has to be repeated from time to time.
- Endpoint or GK may un-register using the URQ message.
- The GK maintains an alias to Network Address translation table.
- Access token may be used later in call setup

Call Admission sets the upper limit
for the aggregate bitrate of the call

Ep \[\text{ARQ}\{\text{admission rq}\} \{\text{Requested} \sum\text{call bandwidth: payload only}\}\]
\[\text{GK}\]
\[\text{ACF}\{\text{may reduce BW, use direct or GK sig}\}/\text{ARJ}\]
[ transport address of call signaling channel]

Call

BRO (Bw Change rq)
Call signalling uses H.225.0

- Call signalling = call setup, request changes in Bw of a call, get status of Ep, disconnect call
- Call signalling is largely inherited from ISDN
- Call Signalling Channel is opened prior to H.245 procedures and prior to any other logical channels between endpoints. Eps have a well known TSAP id for the Call Sig. channel and a well-known Discovery Multicast address.

H.323 Call Signalling Channel Routing

- ACF has the Transport Address of the Call Signalling Channel
- The address is either a GK address or an Endpoint address.

Gatekeeper Routed Call Signalling

Direct Endpoint Call Signalling

Raimo Kantola-S-2004
Signalling Protocols 11 - 21

Raimo Kantola-S-2004
Signalling Protocols 11 - 22
The goal of call signalling is the setup/release of H.245 Control Channel!

**Direct H.245 Control Channel**

1. RAS Signalling Channel msgs
2. Call Signalling Channel msgs
3. H.245 Control Channel

**GK routed H.245 Control**

1. 1,5 - ARQ
2. 2,6 - ACF
3. 7,8 - Connect
4. 9,10 - H.245 Channel

**H.245** carries end-to-end control messages between H.323 entities

- Master/slave determination for conflict resolution
- Capability Exchange (e.g. what codecs are supported)
- Logical Channel Signalling (binds media type, algorithm etc. to ports)
- Bidirectional Logical Channel Signalling
- Close Logical Channel Signalling
- Mode Request (conference modes)
- Round Trip Delay Determination
- Maintenance Loop Signalling
- H.323 also uses flowControlCommand of H.245 to limit bandwidth
Sample H.245 Logical Ch Signalling for two way RTP+RTCP communications setup

In IP networks a logical channel corresponds to an IP port number
- Uses H.245 Control Channel

H.323 Call identification uses

- Call reference value - between two H.323 entities on a signalling channel (one for call signalling and another for RAS channel)
- Call ID - a globally unique non-zero value created by the calling endpoint passed in all H.225 messages
- Conference ID (CID) - in all sub-calls of a conference
Both endpoints registered - Direct/GK-routed call signalling

Ep1                        GK1                                 GK2                      Ep2

ARQ

ACF

Setup

Call Proceeding

Facility

Release Complete

Disengage bw, mode

Setup

Call Proceeding

Alerting

Connect

Gateway decomposition

DSS1 or ISUP  Media Gateway  IP based signaling (e.g H.323)

+ H.248 = Megaco

PCM voice  Media Gateway  RTP + RTCP flow

MG - Trunk gateway, residential gateway etc. Many MGs can be controlled by one MGC, MGCs can be a mated pair --> higher availability performance.
H.323 summary

- H.323 inherits call signaling from ISDN
- H.323 has many conference modes and many signalling and call routing options
- Call setup delay is reduced by using the Fast Connect Procedure: packs all setup info from both H.225.0 and H.245 into fastStart element in setup and connect (call proceeding, alerting) messages
- Versions 1, 2, 3 and 4 are available! Version 4 products are available. Supports HTTP based 3rd party service control. ITU-T has version 5 of H.323 (quick browsing did not reveal anything major new stuff …)
- In conferencing applications over IP H.323 is still the leader.
- Version Interoperability and Vendor interoperability are issues!
- More info e.g. in http://www.h323forum.org/